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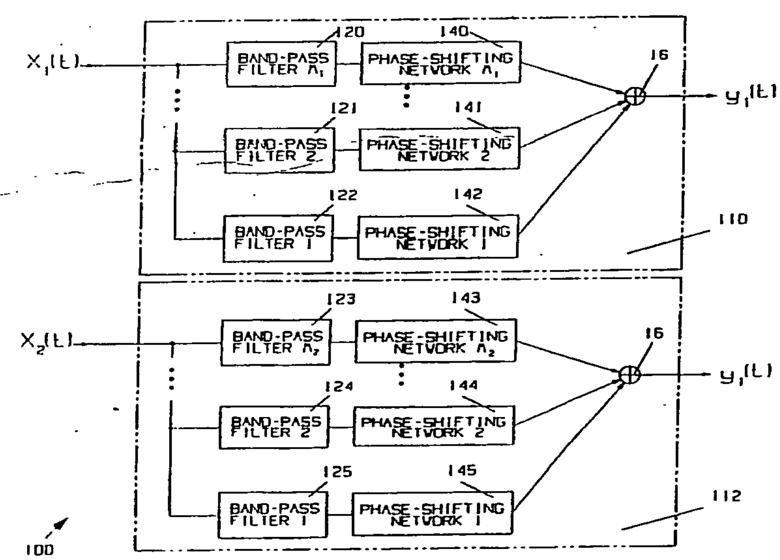
(54) Title: METHOD FOR ELIMINATING THE PRECEDENCE EFFECT IN STEREOPHONIC SOUND SYSTEMS AND RECORDING MADE WITH SAID METHOD



(57) Abstract

A method for eliminating image shift and the precedence effect in stereophonic recordings is disclosed. The preferred embodiment of the invention utilizes random phase shifting of the signals (140-145) in various frequency bands to reduce the cross-correlation between the stereophonic signals.

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METHOD FOR ELIMINATING THE PRECEDENCE EFFECT IN STEREOPHONIC SOUND SYSTEMS AND RECORDING MADE WITH SAID METHOD

Background of the Invention

The present invention relates to the field of acoustic reproduction and, more particularly, to the stereophonic reproduction of an acoustic image through two or more speakers.

When a pair of loudspeakers are driven by stereophonic signals, the perceived locations of the various sound sources generated by the stereophonic signals create for the listener what is known as an acoustic image, i.e., a map of the imaginary physical locations of these sound sources. The apparent location of the sound source is largely determined by the difference in arrival time and the intensity of the relevant component signals generated in the left and right speakers.

Thus, if a particular signal component is fed to both speakers with no relative delay and the same signal amplitude, the component of the acoustic image created by that signal will appear to be located on a line centered between the two speakers. If that signal component arrives fractionally earlier from the left speaker than from the right and/or the intensity of the component from the left speaker is greater than that from the right speaker, its image component should appear to be located left of center. The apparent locations of a set of such image components makes up the composite acoustic image perceived by the listener.

In typical listening environments such as living rooms or theaters, most listeners are located nearer to one loudspeaker than to the other(s). For the

purposes of the following discussion, it will be assumed that the acoustic image is being produced by only two loudspeakers. If the listener moves nearer to one loudspeaker, the sound from that speaker is more intense and reaches the listener ahead of the sound generated at the same time in the other speaker. Hence, moving the listener closer to one speaker is equivalent to introducing an intensity loss and a time delay into the material being reproduced in the other speaker. When very similar material is reproduced by two or more loudspeakers, listeners report that the sound images they perceive are either shifted toward the location of the nearest loudspeaker or almost entirely located in the nearest loudspeaker, depending upon the delay in question.

It should be noted that when a listener moves nearer to one speaker, both the intensity of the sound and the time delay are affected. However, it as been shown that the arrival time difference has a more pronounced and important influence than does the intensity difference.

If the time delay is less than approximately 1.0 msec., listeners describe hearing a single sound image located between the speakers, but shifted toward the closer speaker. This effect is referred to as image shift. If the time delay is greater than approximately 1.0 msec but less than an upper limit discussed below, the listener perceives a single sound image that is located at the closer loudspeaker. The traditional explanation for this phenomenon is that the listener's auditory system has attempted to suppress the delayed signal. This phenomenon is often referred to as the precedence effect, the Haas effect, or the law of the first wavefront. In the following discussion, the effect will be referred to as the precedence effect.

There is an upper limit to the time delay at which the precedence operates. At time delays greater than this limit, the delayed sound is heard. The exact magnitude of this upper limit depends upon the qualities of the sound source. The precedence effect is more pronounced for transient sound sources such as struck or plucked musical instruments than it is for continuous sound sources such as blown or bowed musical instruments. The upper limit is found experimentally to vary from 8 to 70 msec with a typical limit being about 15 msec.

When the precedence effect releases, listeners report that the sound image is located in two loudspeakers. When the loudspeakers are separated by a sufficiently great distance, listeners report hearing two sound images, one of which being echo-like. As the time delay from the difference in distances to the two loudspeakers increases further, the intensity difference also increases significantly. When the intensity difference is approximately 15 dB, the more distant loudspeaker becomes difficult to hear. At this point, listeners report that the sound image is located in one loudspeaker.

Broadly, it is an object of the present invention to provide an improved apparatus and method for stereophonic sound reproduction.

It is another object of the present invention to provide a stereophonic sound reproduction system which generates an acoustic image whose perceived location is independent of the position of the listener relative to the two loudspeakers.

It is a further object of the present invention to provide a stereophonic sound reproduction system which overcomes or eliminates image shift and the precedence effect.

These and other objects of the present invention will become apparent to those skilled in the art from the following detailed description of the invention and the accompanying drawings.

Brief Description of the Drawings

Figure 1 is a block diagram of an apparatus according to the present invention for converting a pair of standard stereophonic input signals into a pair of stereophonic output signals.

Figure 2 illustrates the improvement obtained with the present invention.

Summary of the Invention

The present invention comprises an apparatus for processing first and second input signals to produce first and second stereophonic sound output signals. The apparatus includes a means for receiving said first input signal and a first processing circuit for generating a sum signal having a value substantially equal to the sum of N_1 band-limited signals. The ith said band-limited signal has an amplitude substantially equal to that of the first input signal in a predetermined frequency range ${}^1f_i \pm {}^1\delta f_i$ and a phase which differs from the phase of said input signal in said predetermined frequency range by an amount ${}^1\phi_i$. Here, i runs from 1 to N_1, $N_1 > 2$ and the set of ${}^1\phi_i$ are distributed between $-\pi$ and $+\pi$. The first processing means also includes circuitry for generating the first stereophonic output signal from said sum signal.

The apparatus also includes a means for receiving said second input signal; and a second processing means for generating a sum signal having a value substantially equal to the sum of N_2 band-limited signals. The ith said band-limited signal has an amplitude substantially equal to that of said second input signal in a predetermined frequency range ${}^2f_i \pm {}^2\delta f_i$ and a phase which differs from the phase of said input signal in said predetermined frequency range by an amount ${}^2\phi_i$. Here, i runs from 1 to N_2; $N_2 > 2$, and the set of ${}^2\phi_i$ are distributed between $-\pi$ and $+\pi$. The second processing means includes circuitry for generating the second stereophonic output signal from the sum signal. Here, ${}^1\phi_i$ differs from ${}^2\phi_i$ for at least one value of i.

Detailed Description of the Invention

The present invention comprises a method, and the audio recordings made with said method. The method provides a means for providing improved stereophonic sound reproduction which is capable of generating an acoustic image whose perceived location is substantially independent of the position of the listener relative to the loudspeakers. The system of the present invention utilizes

an apparatus which generates a pair of stereophonic output signals having a cross-correlation which is very close to zero from a pair of standard stereophonic input signals.

The apparatus of the present invention operates by manipulation of the phase relationships of the output signals while maintaining a constant magnitude across frequency. The maintenance of a constant magnitude across frequency avoids changes in the colorization of the output signals. The manipulation of the phase relationships creates a phase incoherence which is sufficient to control the cross-correlation of the output signals. It has been found experimentally that both image shift and the precedence effect are substantially reduced when the cross-correlation measure defined below is between -0.5 and 0.5.

The cross-correlation of two signals, $y_1(t)$ and $y_2(t)$, will be measured, for the purposes of this discussion, in terms of a cross-correlation measure which is defined to be the extreme value of the cross-correlation function $\Omega(\tau)$, where:

$$\Omega(\tau) = \lim_{T->\infty} 1/(2T) \int y_1(t)\, y_2(t+\tau)\, dt \qquad (1)$$

The cross-correlation measure has a maximum possible value of 1 and a minimum possible value of -1. It has been found experimentally that both image shift and the precedence effect are substantially reduced when the cross-correlation measure has an absolute value less than 0.5.

An apparatus for altering the correlation of two signals by adjusting the relative phase as a function of frequency of the two signals is shown in Figure 1. Figure 1 is a block diagram of an apparatus 100 for creating a pair of processed stereophonic output signals, $y_1(t)$ and $y_2(t)$, from a pair of stereophonic input signals, $x_1(t)$ and $x_2(t)$. Output signal $y_1(t)$ is generated by a first channel 110, and output signal $y_2(t)$ is generated by a second channel 112. Each channel generates an output signal by dividing the appropriate input signal into a plurality of components, each component representing the intensity of the signal in a specified frequency band. Apparatus 100 utilizes a plurality of band-pass fil-

ters. Exemplary filters are shown at 120-125. The signal in the ith frequency band is then phase shifted by an amount ϕ_i utilizing a phase shifting network. Exemplary phase shift networks are shown at 140-145. The phase-shifted signals are then summed by signal adder 16 to form the output signal in question.

The number of frequency bands into which each of the input signals is split is preferably the same. That is, $M_1=M_2$. However, it will be apparent to those skilled in the art that different numbers of bands can be used in the different channels provided the frequency bands are chosen as described below.

The cross-correlation measure of the output signals, $y_1(t)$ and $y_2(t)$, is determined by the phase shifts ϕ_i that are added to the various frequency components of $x_1(t)$ and $x_2(t)$. In the preferred embodiment of the present invention, the ϕ_i are chosen randomly between the specified limits of π and $-\pi$. It will be apparent to those skilled in the art that a sequency between P and $P+2\pi$, where P is any constant, will provide identical results. The manner in which the phase shifts ϕ_i are chosen between the specified limits is important in determining the quality of the output signals. In the preferred embodiment of the present invention, the ϕ_i are chosen by generating a sequence of random numbers between π and $-\pi$. Although the preferred embodiment of the present invention utilizes randomly selected phase shifts, other methods of selecting the phase shifts in question may be utilized without departing from the teachings of the present invention. The important factor here is that the absolute value of the cross-correlation measure of y_1 and y_2 be made less than approximately 0.5.

The phase shifts added to the band-limited signals in the first channel 110 should be different than those added to the band-limited signals in the second channel 112. It will be apparent to those skilled in the art that the apparatus of the present invention will still function satisfactorily if a small percentage of the phase shifts in the first channel 110 are identical to the corresponding phase shifts in the second channel 112.

In order to avoid the perception of a banded or broken image, the preferred embodiment of the present invention satisfies additional constraints.

First, the phase shifts selected should change rapidly from band to band. In other words, on average, the change in phase angle between adjacent pairs of phase shifts, ϕ_i, ϕ_{i+1}, and ϕ_{i+2}, should not be monotonic; however, due to the statistics inherent in random selection, this requirement will not always be met.

The size of the frequency bands utilized in the present invention also affects the performance of the invention. It is found experimentally that frequency bands as small as 50Hz perform satisfactorily. In fact, it is observed that smaller bandwidths perform better than larger bandwidths. The factors which determine the smallest bandwidth will discussed in more detail below.

The above described embodiments of the present invention utilize bandpass filters and phase-shift circuits. The same results may be obtained, however, by convolving $x_1(t)$ and $x_2(t)$ with a filter functions $h_1(t)$ and $h_2(t)$ to produce $y_1(t)$ and $y_2(t)$. That is:

$$y_1(t) = \int x_1(t-z)h_1(z)dz \tag{2}$$

$$y_2(t) = \int x_2(t-z)h_2(z)dz \tag{3}$$

The transformation functions $h_1(t)$ and $h_2(t)$ provide a phase-shifting of the individual frequency bands.

The present invention preferably utilizes digital input signals. If the signal source consists of analog signals, they may be converted to digital form via a conventional analog-to-digital converters. In this case, each output signal consists of a sequence of digital values. The ith value for each output signal corresponds to the value of the output signal at time iT, where T is the time between digital samples. In this case, the convolution operation given in Eq. (2) reduces to:

$$y_1(nT) = y_n = \Sigma_m \; x_{n-m} \, {}^1h_m \tag{4}$$

where m runs from 0 to N-1. The filter coefficients, h_m are calculated from:

$$^1h_m = (1/N_1)\ \Sigma_k\ \exp(kmw + {}^1p_k) \qquad (5)$$

Here, k runs from 0 to N_1-1, $w=2*\pi/N_1$, $\exp(z)=e^{jz}$, and N_1 is the total number of frequency samples. A similar convolution operation is used to generate the second output signal, $y_2(nT)$, with the exception that a different set of ^2p_k are used.

The number of frequency samples, N_1 or N_2 directly specified in the frequency domain and used to create the incoherent time domain signal is limited by the number of points comprising the time domain signal. Typically, these points are linearly spaced across frequency. The filter coefficients that result from using the Fast Fourier Transform given in Eq. (5) will not be constant between the specified frequency points. As a result, timbral neutrality will be completely achieved only if this number is very large in the above described equations. There is a practical limit to the size of this number in commercially realizable apparatuses.

In addition, for complete timbral neutrality, the integrals given in Eqs. (1) and (2) must be performed from $-\infty$ to $+\infty$. However, in practice, the maximum acceptable convolution time is of the order to 20 msec. If longer times are chosen, transient properties of the input signal are smeared in time. Hence, for any given sampling rate, there is a trade-off between timbral neutrality and the effect at low frequencies. Hence, the smallest frequency bands into which each stereophonic input signal can be satisfactorily broken is of the order of 50HZ.

The present invention minimizes the effects of this trade-off by selecting the particular random number sequence used in generating the phase shifts. It has been found experimentally that different sets of phase shifts, $^i\phi_k$, produce different subjective effects on the listener. Hence, in the preferred embodiment of the present invention, a number of different sets of phase shifts are generated, and the set producing the best effect, as judged by listening to the output signals,

is utilized.

The output signals provided by the present invention may be played through conventional speakers. These signals may also be recorded onto conventional stereophonic recording media for subsequent playback through conventional stereophonic equipment. In other words, a pair of traditional stereophonic input signals may be converted to stereophonic output signals, using the apparatus of the present invention, and these stereophonic output signals may then be recorded on the desired storage medium, to be reproduced later by playing the signals back through a conventional stereo sound reproduction system.

A recording according to the present invention would have first and second channels in which the signal in the first channel had an intensity in each of a M predetermined frequency bands $f_i \pm \delta f_i$ that was proportional to the intensity of the second channel in said predetermined frequency bands. Here, i runs from 1 to M. However, the phase of the signal in the first channel in each of said predetermined frequency bands would differ from that of the second channel signal in the corresponding frequency band by ϕ_i. The ϕ_i are preferably a random sequence between $-\pi$ and $+\pi$. However, any sequence which results in the first and second channels having proportional amplitudes as a function of frequency and a cross-correlation measure less than 0.5 is acceptable.

An apparatus according to the present invention may be installed before the loudspeaker in each channel of a conventional stereophonic sound reproduction system to perform the phase conversion on-line. It will be apparent to those skilled in the art that such an apparatus converts the original signal in each channel to a new signal having an intensity in each of M predetermined frequency bands $f_i \pm \delta f_i$ which is proportional to the original signal in said frequency bands. The phase of the new signal in each of said would differ from that of the orignial signal by ϕ_i, where i runs from 1 to M. The ϕ_i are chosen such that the cross-correlation measure of the original signal and the new signal is less than 0.5.

The improvements obtained with the present invention are illustrated in Figure 2. Figure 2 compares the subjective response of a number of subjects to stereophonic program material being played through two speakers. The x-axis of the graph is the time delay between the signals from the two speakers. Each observer is asked to classify the sound source as coming from either one speaker or more than one speaker. As can be seen from the Figure, the present invention substantially reduces the precedence effect.

The above description has utilized a two channel stereophonic system for purposes of illustration. However, it will be apparent to those skilled in the art that the principles of the present invention may be applied to stereophonic systems having more than two channels. In this case, each channel is assigned a separate set of phase shifts and the signals in that channel processed as described above. In the preferred embodiment of the present invention, different sets of randomly chosen phase shifts are utilized. However, other sets of phase shifts may be utilized if the resultant output signals have pairwise cross-correlation measures which are less than 0.5.

It will also be apparent to those skilled in the art that not all of the channels need be processed. In particular, the processing circuitry in one of the channels may be replaced by a delay circuit having a delay equal to half the processing delay in the other channels. For example, channel 112 shown in Figure 1 may be replaced by a delay circuit having a delay equal to one half the processing delay of channel 110. It has been found, experimentally, that replacing one channel by a delay circuit improves the perceived timbral fidelity of the output signals. The replacement of channel 112 by a delay circuit is functionally equivalent to setting each ${}^{2}\phi_i$ to zero for $i=1$ to M_2.

There has been described herein a novel apparatus and method for defeating precedence effects. Although the above embodiments of the present invention have been described with reference to a two channel stereophonic system, it will be apparent to those skilled in the art that the principles described above may be utilized in systems having more than two channels. In this case, each channel is convolved with a different set of random phase shifts.

It will also be apparent to those skilled in the art that the stereophonic output signals produced by the present invention can be recorded on any suitable medium and then played back through a conventional stereophonic system.

Various modifications of the present invention will become apparent to those skilled in the art from the foregoing description and accompanying drawings. Accordingly, the present invention is to be limited solely by the scope of the following claims.

WHAT IS CLAIMED IS:

1. A method for reducing precedence effect in a sterophonic sound system comprising first and second sound tracks, said method comprising processing said first sound track to provide a new first sound track, said new first sound track having an intensity as a function of frequency which is substantially proportional to said first sound track, wherein the absolute value of the cross-correlation measure of said first sound track and said new first sound track is less than 0.5.

2. The method of Claim 1 wherein said processing step comprises generating a sum signal having a value substantially equal to the sum of M_1 band-limited signals, the ith said band-limited signal having an amplitude substantially equal to that of said first sound track in a predetermined frequency range $f_i \pm \delta f_i$ and a phase which differs from the phase of said first sound track in the ith said predetermined frequency range by an amount ϕ_i, i running from 1 to N_1.

3. The method of Claim 2 wherein said ϕ_i are a random sequence of values between $-\pi$ and $+\pi$.

4. The method of Claim 2 further comprising the step of delaying said second sound track by a predetermined time to compensate for the time needed to perform said processing step on said first sound track.

5. The method of Claim 2 wherein said first sound track comprises a sequence of digital values and wherein said processing step comprises convolving said sequence of digital values with a set of filter coefficients.

6. A recording comprising first and second sound tracks wherein said first sound track intensity as a function of frequency is proportional to the intensity of said second sound tracks and said first and second sound tracks have an absolute value of the cross-correlation measure less than 0.5.

7. The recording of Claim 6 wherein said first sound track comprises a

sum signal having a value substantially equal to the sum of M_1 band-limited signals, the ith said band-limited signal having an amplitude substantially equal to that of said secondsound track in a predetermined frequency range $f_i \pm \delta f_i$ and a phase which differs from the phase of said second sound track in the ith said predetermined frequency range by an amount ϕ_i, i running from 1 to M.

8. The recording of Claim 7 wherein said ϕ_i comprises a random sequence of values between $-\pi$ and $+\pi$.

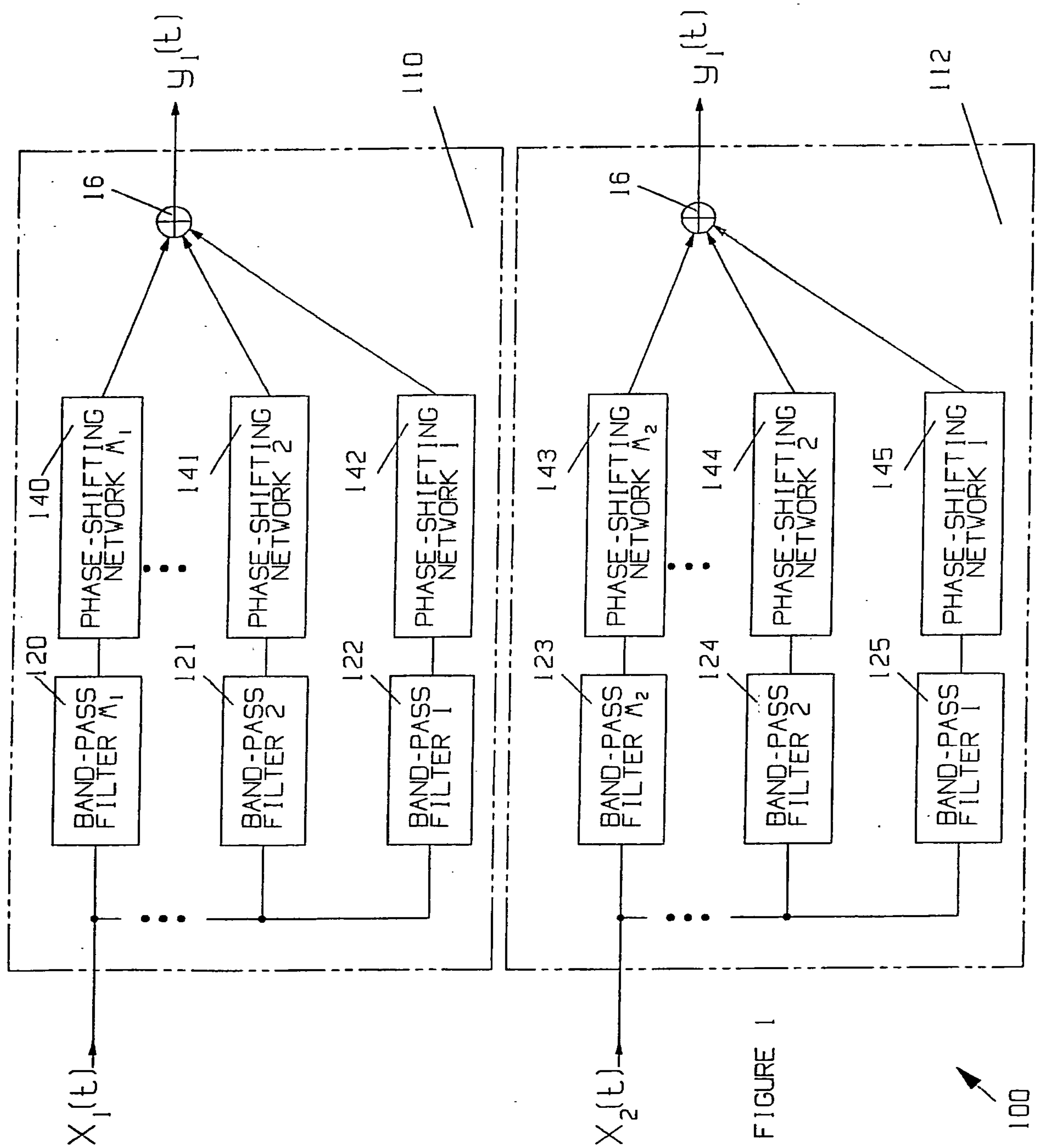

FIGURE 1

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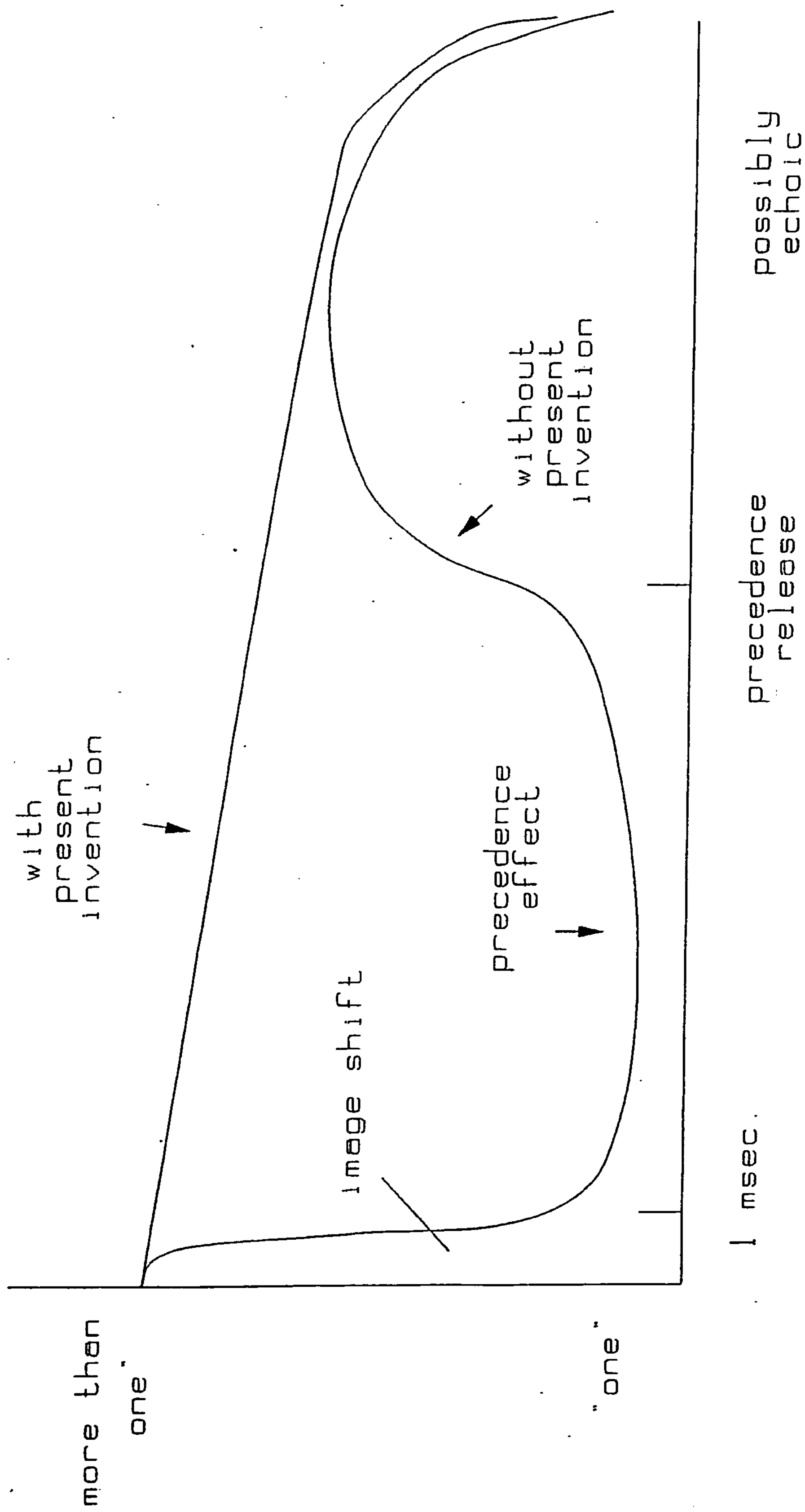

FIGURE 2

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INTERNATIONAL SEARCH REPORT

International Application No PCT/US91/04168

I. CLASSIFICATION OF SUBJECT MATTER (if several classification symbols apply, indicate all) [3]

According to International Patent Classification (IPC) or to both National Classification and IPC
IPC(5): H04R 5/00; A61F 11/06; H03B 29/00
U.S.A. Cl. 381/1,61,63,17,71

II. FIELDS SEARCHED

Minimum Documentation Searched [4]

Classification System	Classification Symbols
U.S.	381/1,61,63,17,71

Documentation Searched other than Minimum Documentation to the Extent that such Documents are Included in the Fields Searched [5]

III. DOCUMENTS CONSIDERED TO BE RELEVANT [14]

Category *	Citation of Document, [16] with indication, where appropriate, of the relevant passages [17]	Relevant to Claim No. [18]
X	US, A, 4,121,059 (NAKABAYASHI) 17 October 1978 See Fig. 6 and Fig. 8, col. 6, lines 3-12.	1-8
Y	US, A, 4,423,289 (SWINBANKS) 27 December 1973 See Fig. 2 and Fig. 5	1-8
Y	US, A, 2,776,020 (W.B. CONOVER et al) 01 January 1957 See Fig. 2	1-8
A	US, A, 2,183,248 (R. R. RIESZ) 12 December 1939	1-8

* Special categories of cited documents: [15]

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IV. CERTIFICATION

Date of the Actual Completion of the International Search [2]	Date of Mailing of this International Search Report [2]
08 August 1991	17 SEP 1991
International Searching Authority [1]	**Signature of Authorized Officer [20]**
ISA/US	Jack Chiang